REMARKS

In view of the following discussion, the Applicants submit that none of the claims now pending in the application are unpatentable under the provisions of 35 U.S.C. § 103. The Applicants herein amend claims 1 and 16. Support for the amendments may be found in the Applicants' specification on at least paragraph [18]. Thus, the Applicants believe that all of these claims are now in allowable form.

I. REJECTION OF CLAIMS 1-4, 7-19 AND 23-31 UNDER 35 U.S.C. § 103

A. Claims 1-4, 7-11, 16-19, and 23-27

The Examiner rejected claims 1-4, 7-11, 16-19, and 23-27 as being unpatentable over Summers, et al. (U.S. Patent No. 6,961,416, issued on November 1, 2005, hereinafter referred to as "Summers") in view of Rodman, et al. (U.S. Patent Publication No. 2002/0103864, published on August 1, 2002, hereinafter referred to as "Rodman") and in further view of Aravamudan, et al. (U.S. Patent No. 6,584,076, issued on June 24, 2003, hereinafter referred to as "Aravamudan"). The Applicants note that the Examiner included claims 20-22 in the rejection. However, the Applicants note that claims 20-22 were previously canceled without prejudice. Thus, the Applicants respectfully traverse the rejection.

Summers teaches an internet-enabled conferencing system and method accommodating PSTN and IP traffic. A caller may call into a conference call by dialing a number connecting them to a Voice node or VoIP node within a chassis on a TDM bus. (See Summers, col. 11, ll. 26-65).

Rodman teaches a system and method for coordinating a conference using a dedicated server. The system and method initiates a data conference between a plurality of conference endpoints linked in communication by a private or public computer network. (See Rodman, Abstract).

Aravamudan teaches a telecommunications conferencing method and apparatus. The method and apparatus use a plurality of device servers including a packet circuit gateway. In response to a request for a conference call, the

packet network determines the parties to be on the conference call and selects a conference bridge that results in the lowest cost for the conference call. (See Aravamudan, Abstract).

The Examiner's attention is directed to the fact that Summers, Rodman and Aravamudan, alone or in any permissible combination, fail to teach or to suggest a method or apparatus for establishing a VoIP conference call comprising <u>receiving an indication at a Voice Conference Server (VCS) from the first VoIP station in the private network for joining one of a plurality of existing conversations on said VCS between the plurality of communication stations, wherein said plurality of existing conversations are VoIP calls and each one of said plurality of existing conversations is on a different connection</u> and <u>identifying said one of said plurality of existing conversations on said VCS via a code number entered by said first VoIP station corresponding to the second VoIP station</u>, as positively claimed by the Applicants' independent claims 1 and 16, respectively. Specifically, Applicants' independent claims 1 and 16 recite:

> 1. A method for establishing a Voice over Internet Protocol (VoIP) conference call by joining a first VoIP station in a communication between a plurality of communication stations, wherein at least one of the plurality of communication stations is a second VoIP station in a private network and said first VoIP station is in the private network, the method comprising:
>
> <u>receiving an indication at a Voice Conference Server (VCS) from the first VoIP station in the private network for joining one of a plurality of existing conversations on said VCS between the plurality of communication stations, wherein said plurality of existing conversations are VoIP calls and each one of said plurality of existing conversations is on a different connection</u>;
>
> <u>identifying said one of said plurality of existing conversations on said VCS via a code number entered by said first VoIP station corresponding to the second VoIP station</u>, wherein said one of said plurality of existing conversations is between the second VoIP station in the private network and a phone in a public network, wherein said VCS is external to said first VoIP station and said plurality of communication stations;
>
> establishing a Real-Time Transport Protocol (RTP) voice path with the first VoIP station and said VCS; and

> managing data packet transmission between the first VoIP station and one of the plurality of communication stations via said VCS. (Emphasis added).
>
> 16. A device for establishing a Voice over Internet Protocol (VoIP) conference call by joining a first VoIP station in a communication between a plurality of communication stations, wherein at least one of the plurality of communication stations is a second VoIP station in a private network and said first VoIP station is in the private network, the device comprising:
>
> a receiver in a Voice Conference Server (VCS) for receiving an indication from the first VoIP station in the private network for joining one of a plurality of existing conversations on said VCS, wherein each one of said plurality of existing conversations is on a different connection, wherein said indication comprises a code number entered by said first VoIP station corresponding to the second VoIP station identifying said one of said plurality of existing conversations on said VCS, wherein said one of said plurality of existing conversations is between the second VoIP station in the private network and a phone in a public network, wherein said VCS is external to said first VoIP station and said plurality of communication stations;
>
> an apparatus in said VCS for setting up a Real-Time Transport Protocol (RTP) voice path with the first VoIP station in response to the received signal for joining said call; and,
>
> an RTP mixer in said VCS for managing at least two VoIP stations and sending the mixed data packets to at least one VoIP station. (Emphasis added).

In one embodiment, Applicants' invention is a method or apparatus for establishing a VoIP conference call comprising <u>receiving an indication at a Voice Conference Server (VCS) from the first VoIP station in the private network for joining one of a plurality of existing conversations on said VCS between the plurality of communication stations, wherein said plurality of existing conversations are VoIP calls and each one of said plurality of existing conversations is on a different connection</u> and <u>identifying said one of said plurality of existing conversations on said VCS via a code number entered by said first VoIP station corresponding to the second VoIP station</u>. As a result, the Applicants' invention provides conferencing capability in private VoIP networks while containing costs for the VoIP phones because of the VCS. (See e.g., Applicants' specification, p. 6, para. [15]). Moreover, the VCS may provide

conferencing capabilities without the need to pre-establish a conference call. (See *Id.* para. [16]).

The alleged combination (as taught by Summers) fails to teach or suggest a method or apparatus for establishing a VoIP conference call comprising <u>receiving an indication at a Voice Conference Server (VCS) from the first VoIP station in the private network for joining one of a plurality of existing conversations on said VCS between the plurality of communication stations, wherein said plurality of existing conversations are VoIP calls and each one of said plurality of existing conversations is on a different connection</u> and <u>identifying said one of said plurality of existing conversations on said VCS via a code number entered by said first VoIP station corresponding to the second VoIP station</u>. The Applicants first note that the Applicants' invention does not require a user to pre-establish a conference call and distribute conference call information to other participants. In other words, a user simply sends an indication to join an existing conversation, identifies the call they wish to join and the VCS connects the user to the existing conversation.

Notably, all of the references cited by the Examiner only teach traditional methods of conference calling. That is, each reference cited by the Examiner <u>requires a conference call to be pre-established</u> and each participant to dial into a conference call number that is assigned to the pre-established conference call. This significant difference is clear upon examination of the claim language recited in the Applicants' independent claims 1 and 16.

Summers clearly teaches that a conference is scheduled and that once the scheduled start time arrives that callers join the conference using a dial in or dial out procedure. (See Summers, col. 11, ll. 26-31). For example, the caller dials a conference telephone number or IP address <u>previously provided to the caller</u>. (See *Id.* at ll. 36-40, emphasis added).

In stark contrast, the Applicants' invention does not require a user to dial a conference call number. Rather, the user in the Applicants' invention simply sends an indication to the VCS that they desire to join an <u>existing conversation</u>.

In addition, Summers teaches away from the Applicants' invention because Summers teaches that each caller must dial into the same conference call telephone number. In other words, Summers teaches that all callers are on the same connection.

In stark contrast, the Applicants' invention teaches receiving an indication at a Voice Conference Server (VCS) from the first VoIP station in the private network for joining one of a plurality of existing conversations on said VCS between the plurality of communication stations, wherein said plurality of existing conversations are VoIP calls and each one of said plurality of existing conversations is on a different connection. In other words, multiple communication stations may carry a conversation via the VCS to allow a user to join any one of the existing conversations. To do so, the user simply needs to identify which connection the user wishes to join via the VCS.

Moreover, Summers fails to teach or suggest identifying said one of said plurality of existing conversations on said VCS via a code number entered by said first VoIP station corresponding to the second VoIP station. At best, the conference call numbers taught by Summers identify a pre-established conference call and not a second VoIP station. In addition, the Examiner concedes that Summers fails to teach or suggest this limitation. (See Final Office Action, p 3, ll. 14-20).

In addition, Rodman fails to bridge the substantial gap left by Summers because Rodman also fails to teach or suggest receiving an indication at a Voice Conference Server (VCS) from the first VoIP station in the private network for joining one of a plurality of existing conversations on said VCS between the plurality of communication stations, wherein said plurality of existing conversations are VoIP calls and each one of said plurality of existing conversations is on a different connection and identifying said one of said plurality of existing conversations on said VCS via a code number entered by said first VoIP station corresponding to the second VoIP station. As previously noted, Rodman teaches away from the Applicants' invention because Rodman teaches that the initial audio conference is established by dialing a telephone

number and entering a code assigned to the conference bridge. (See Rodman, para. [0038]). The codes referred to by the Examiner are used with a data conference that may be initiated only for those participants already in the existing audio conference. (See Rodman, para. [0038]-[0039]).

Moreover, the data conference codes used in Rodman are for allowing an invited participant to join a newly generated data conference. (See Rodman, para. [0047]). Notably, the data conference code does not identify an existing conversation between the second VoIP station in the private network and a phone in a public network.

The Examiner continues to cite paragraphs [0011]-[0013] of Rodman. However, the paragraphs in Rodman cited by the Examiner further support the Applicants' arguments. For example, Rodman on paragraph [0012] teaches that to initialize a conference call, a request is sent to a conference server. Then the conference server generates a conference code that is transmitted to the requesting conference endpoint. (See Rodman, para. [0012]). Notably, no conference call is on-going at this point.

Then, the requesting conference endpoint transmits a conference invitation to remote conference endpoints including the conference code. (See Rodman, para. [0013]). After receiving the invitation, the remote conference endpoints may join the pre-established conference call. (See *Id.*). Thus, all of the callers are on the same conference call (i.e. the same connection). Moreover, all of the "codes" taught by Rodman only identify the pre-established conference call. Notably, none of the "codes" taught by Rodman correspond to the second VoIP station.

Finally, Aravamudan also fails to bridge the substantial gap left by Summers and Rodman because Aravamudan also fails to teach or suggest receiving an indication at a Voice Conference Server (VCS) from the first VoIP station in the private network for joining one of a plurality of existing conversations on said VCS between the plurality of communication stations, wherein said plurality of existing conversations are VoIP calls and each one of said plurality of existing conversations is on a different connection and identifying

said one of said plurality of existing conversations on said VCS via a code number entered by said first VoIP station corresponding to the second VoIP station. Aravamudan only teaches dynamically changing conference bridges. (See Aravamudan, Abstract). Therefore, Summers, Rodman and Aravamudan, alone or in any permissible combination, fail to teach or suggest a method or apparatus for establishing a VoIP conference call comprising receiving an indication at a Voice Conference Server (VCS) from the first VoIP station in the private network for joining one of a plurality of existing conversations on said VCS between the plurality of communication stations, wherein said plurality of existing conversations are VoIP calls and each one of said plurality of existing conversations is on a different connection and identifying said one of said plurality of existing conversations on said VCS via a code number entered by said first VoIP station corresponding to the second VoIP station, as positively recited by Applicants' independent claims 1 and 16. Therefore, the Applicants respectfully request the rejection be withdrawn.

Moreover, dependent claims 2-4, 7-11, 17-19, and 23-27 depend, either directly or indirectly, from independent claims 1 and 16, respectively, and recite additional limitations. As such, and for the exact same reason set forth above, the Applicants submit that claims 2-4, 7-11, 17-19, and 23-27 are also patentable over Summers, Rodman and Aravamudan. As such, the Applicants respectfully request the rejection be withdrawn.

B. Claims 12-15 and 28-31

The Examiner rejected claims 12-15 and 28-31 as being unpatentable over Summers in view of Rodman and Aravamudan and in further view of Canon, et al. (U.S. Patent No. 6,269,159, issued on July 31, 2001, hereinafter referred to as "Cannon"). The Applicants respectfully traverse the rejection.

The teachings of Summers, Rodman and Aravamudan have been discussed above. Cannon teaches conferencing with a calling party. The method and apparatus provides three way conferencing which allows a third party caller to call into an existing telephone call at a single line of a called party's

telephone. (See Cannon, Abstract.)

The Examiner's attention is directed to the fact that the alleged combination (as taught by Summers, Rodman, Aravamudan, and Cannon) fails to disclose the novel a method or apparatus for establishing a VoIP conference call comprising receiving an indication at a Voice Conference Server (VCS) from the first VoIP station in the private network for joining one of a plurality of existing conversations on said VCS between the plurality of communication stations, wherein said plurality of existing conversations are VoIP calls and each one of said plurality of existing conversations is on a different connection and identifying said one of said plurality of existing conversations on said VCS via a code number entered by said first VoIP station corresponding to the second VoIP station, as positively claimed by the Applicants' independent claims 1 and 16. (See *supra*).

The Applicants' invention teaches a method or apparatus for establishing a VoIP conference call comprising receiving an indication at a Voice Conference Server (VCS) from the first VoIP station in the private network for joining one of a plurality of existing conversations on said VCS between the plurality of communication stations, wherein said plurality of existing conversations are VoIP calls and each one of said plurality of existing conversations is on a different connection and identifying said one of said plurality of existing conversations on said VCS via a code number entered by said first VoIP station corresponding to the second VoIP station. In contrast, as discussed above, the combination of Summers, Rodman and Aravamudan simply does not teach or suggest the novel method or apparatus for establishing a VoIP conference call comprising receiving an indication at a Voice Conference Server (VCS) from the first VoIP station in the private network for joining one of a plurality of existing conversations on said VCS between the plurality of communication stations, wherein said plurality of existing conversations are VoIP calls and each one of said plurality of existing conversations is on a different connection and identifying said one of said plurality of existing conversations on said VCS via a code number entered by said first VoIP station corresponding to the second VoIP station.

Moreover, Cannon does not bridge the substantial gap left by Summers, Rodman and Aravamudan because Cannon also fails to teach or suggest a method or apparatus for establishing a VoIP conference call comprising receiving an indication at a Voice Conference Server (VCS) from the first VoIP station in the private network for joining one of a plurality of existing conversations on said VCS between the plurality of communication stations, wherein said plurality of existing conversations are VoIP calls and each one of said plurality of existing conversations is on a different connection and identifying said one of said plurality of existing conversations on said VCS via a code number entered by said first VoIP station corresponding to the second VoIP station. Cannon only teaches a method and apparatus for conferencing with a calling party. (See Cannon, Abstract). Thus, for all of the above reasons, the Applicants respectfully contend that claims 1 and 16 of the present invention are not made obvious by the combination of Summers, Rodman, Aravamudan and Cannon.

Furthermore, dependent claims 12-15 and 28-31 depend, either directly or indirectly, from claims 1 and 16, respectively, and recite additional limitations. As such, and for the exact same reason set forth above, the Applicants submit that claims 12-15 and 28-31 are also patentable and not made obvious by the teachings of Summers, Rodman, Aravamudan and Cannon. As such, the Applicants respectfully request the rejection be withdrawn.

CONCLUSION

Thus, the Applicants submit that all of these claims now fully satisfy the requirements of 35 U.S.C. § 103. Consequently, the Applicants believe that all these claims are presently in condition for allowance. Accordingly, both reconsideration of this application and its swift passage to issue are earnestly solicited.

If, however, the Examiner believes that there are any unresolved issues requiring the maintenance of the present final action in any of the claims now pending in the application, it is requested that the Examiner telephone Mr. Kin-Wah Tong, Esq. at (732) 842-8110 so that appropriate arrangements can be made for resolving such issues as expeditiously as possible.

Respectfully Submitted,

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